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(54) **METHODS AND APPARATUS FOR MEASURING SIGNAL LEVEL AND DELAY AT MULTIPLE SENSORS**

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(52) **U.S. Cl.** **381/94.7**; 381/71.11; 708/322

(58) **Field of Search** 381/94.7, 103, 381/94.1, 73.1, 66, 91, 92, 122, 71.1, 71.11; 708/322

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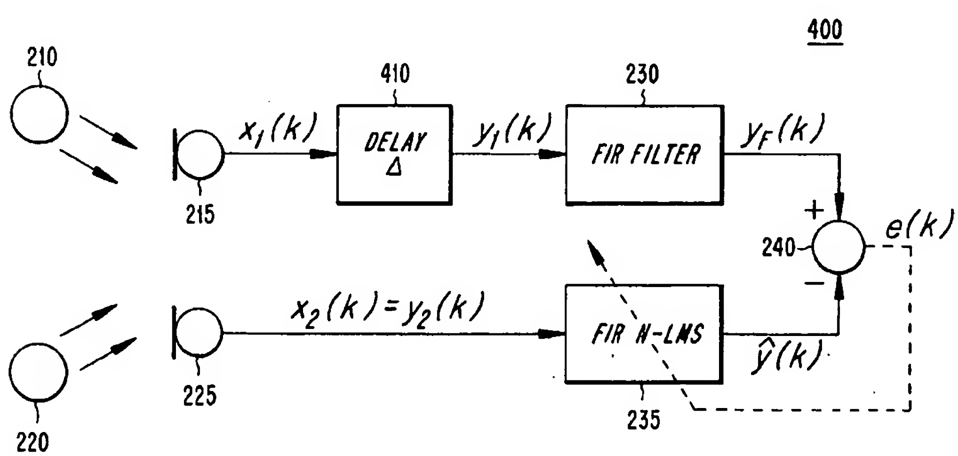
(57) **ABSTRACT**

Methods and apparatus for automatic gain and time delay computations between multiple sensors. In exemplary embodiments, one sensor is treated as a reference sensor with respect to a measured signal quantity which is assumed to be a narrowband process. The relative gain and relative time delay in the additional sensors with respect to the same measured quantity are then automatically calculated based on an adaptive filtering algorithm. Advantageously, the disclosed embodiments are implemented using standard digital signal processing components.

30 Claims, 4 Drawing Sheets



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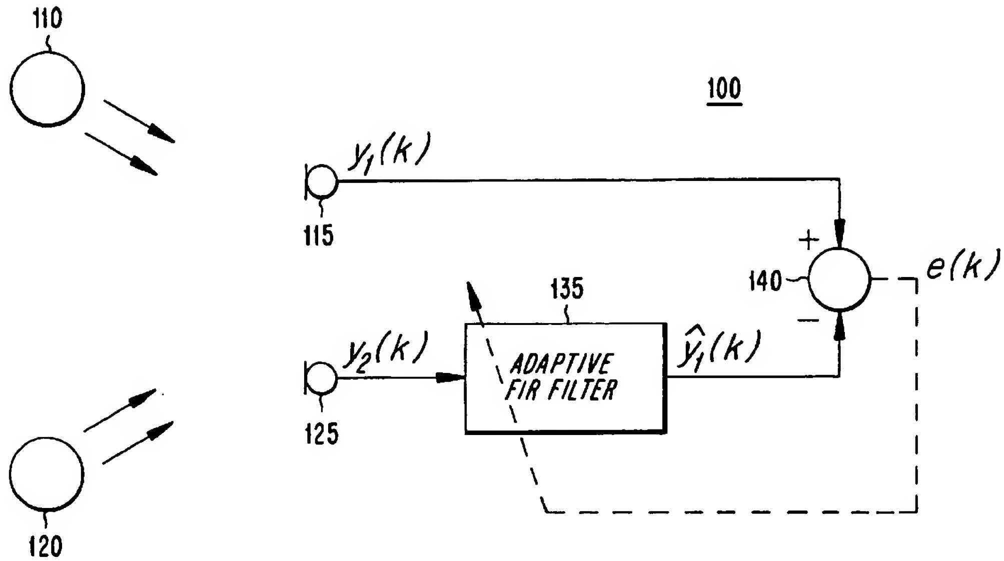

FIG. 1
(PRIOR ART)

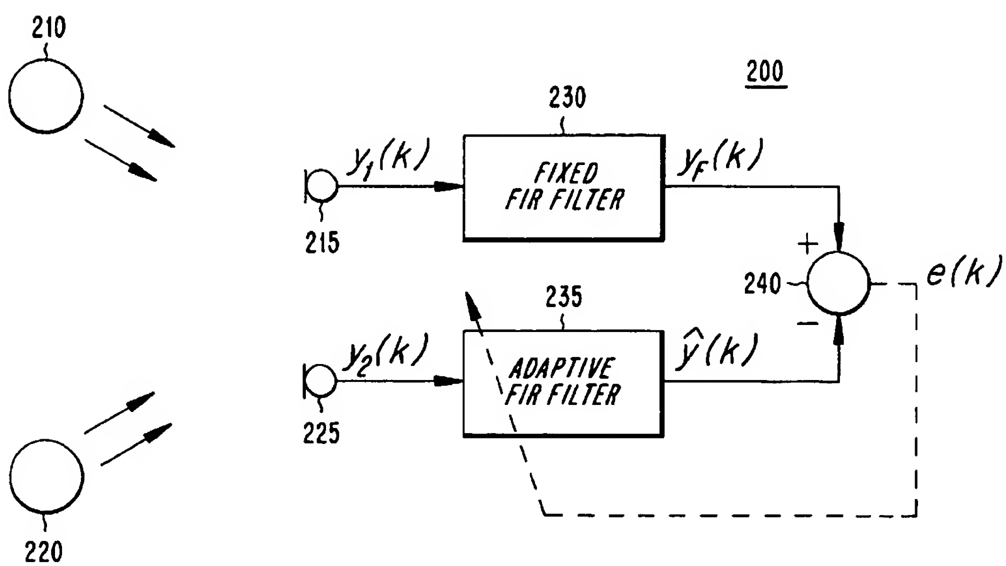

FIG. 2

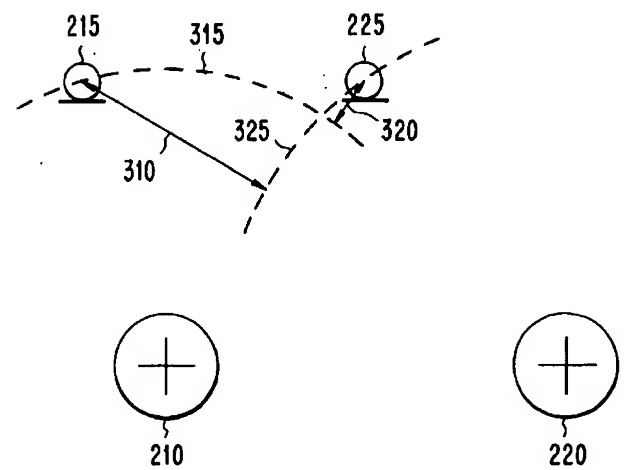
215
315
225
325
320
310
210
220

FIG. 3

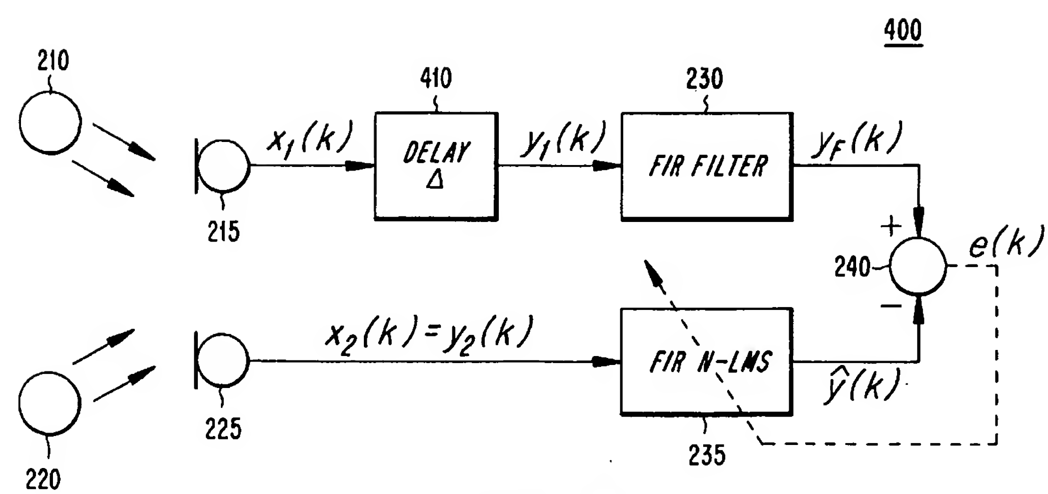
400
210
410
230
$x_1(k)$
DELAY
Δ
$y_1(k)$
FIR FILTER
$y_F(k)$
215
+
240
e(k)
−
$x_2(k)=y_2(k)$
FIR N-LMS
$\hat{y}(k)$
225
235
220

FIG. 4

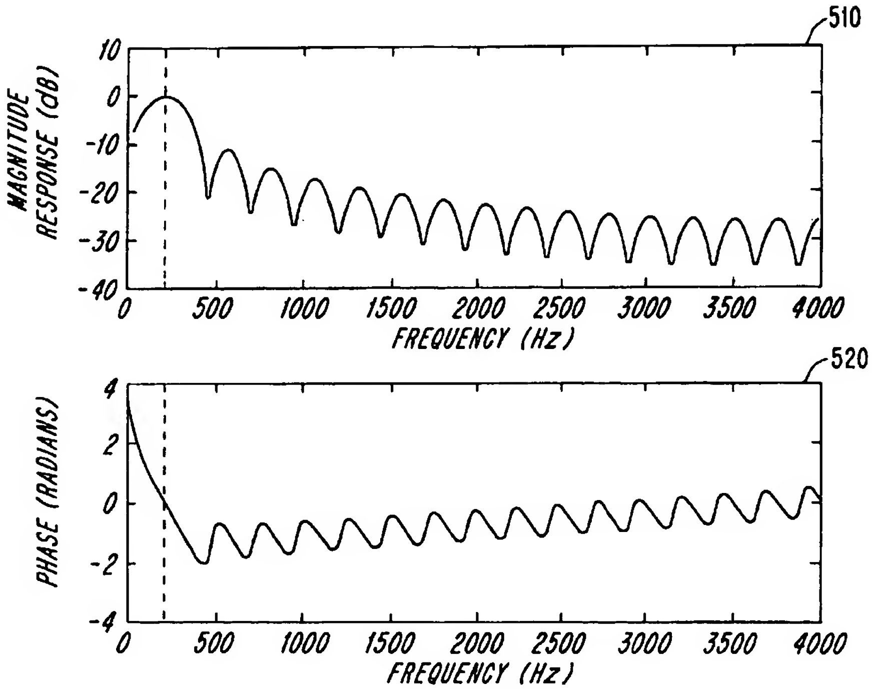
510
10
0
-10
-20
-30
-40
MAGNITUDE RESPONSE (dB)
0
500
1000
1500
2000
2500
3000
3500
4000
FREQUENCY (Hz)
520
4
2
0
-2
-4
PHASE (RADIANS)
FREQUENCY (Hz)

FIG. 5

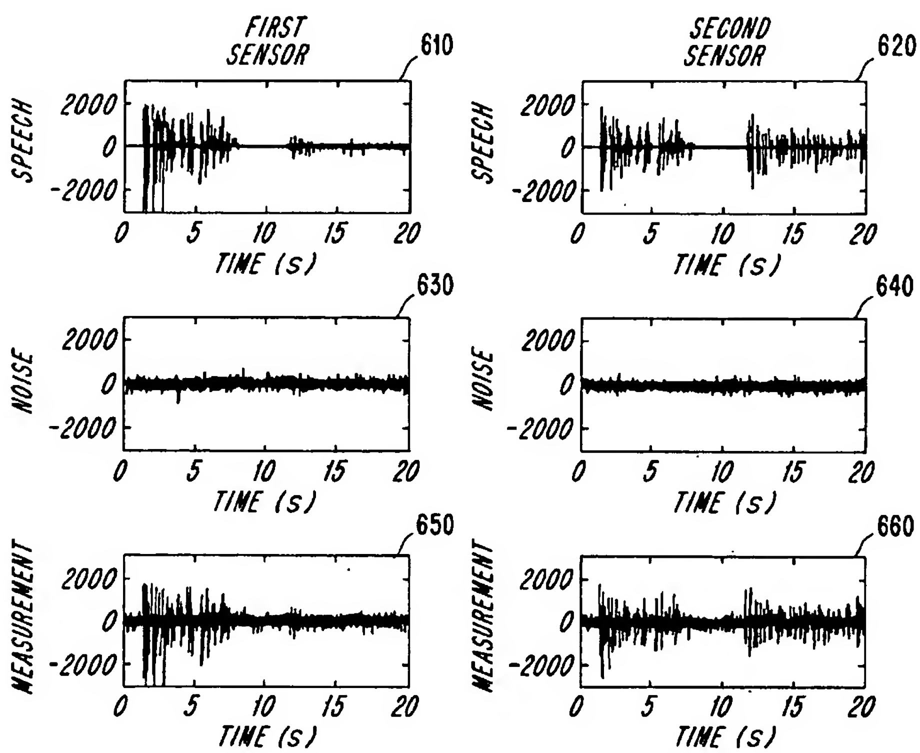
FIRST SENSOR
SECOND SENSOR
610
620
630
640
650
660
SPEECH
NOISE
MEASUREMENT
2000
0
-2000
0
5
10
15
20
TIME (s)

FIG. 6

METHODS AND APPARATUS FOR MEASURING SIGNAL LEVEL AND DELAY AT MULTIPLE SENSORS

BACKGROUND

The present invention relates to signal processing, and more particularly to the measurement of signal levels and time delays at multiple signal sensors.

In many signal processing applications, it is desirable to determine the relative sensitivity of multiple signal sensors with respect to a particular signal source. For example, in the context of hands-free mobile telephony, dual microphones can be used in combination with beamforming methods to reduce the effects of background noise and echoes in an automobile. To do so, information regarding the relative sensitivities of the microphones with respect to different acoustic sources is used, for example, to form a spatial beam toward a particular user and/or to form a spatial notch against another user or a loudspeaker. Such an approach requires that dynamic information with respect to microphone sensitivity be quickly and accurately obtained.

FIG. **1** depicts a prior art system **100** for measuring the relative sensitivities of dual microphones with respect to different signal sources in the context of hands-free mobile telephony. As shown, the prior art system **100** includes a first microphone **115**, a second microphone **125**, an adaptive filter **135** and a summing device **140**. An output $y_1(k)$ of the first microphone **115** is coupled to a positive input of the summing device **140**, and an output $\hat{y}_2(k)$ of the second microphone **125** is coupled to an input of the adaptive filter **135**. An output $\hat{y}_1(k)$ of the adaptive filter **135** is coupled to a negative input of the summing device **140**, and an output e(k) of the summing device **140** is used as a feedback signal to the adaptive filter **135**.

As shown, the first microphone **115** is positioned nearer a first source **110**, and the second microphone **125** is positioned nearer a second source **120**. For example, the first microphone **115** can be a hands-free microphone attached to a sun visor situated nearer a driver of an automobile, and the second microphone **125** can be a built-in microphone within a mobile unit attached nearer a passenger in the automobile. Although it is not shown in FIG. **1**, those skilled in the art will appreciate that analog pre-processing and analog-to-digital conversion circuitry can be included at the output of each of the first and second microphones **115**, **125** so that digital signals are processed by the adaptive filter **135** and the summing device **140**. The output e(k) of the summing device **140** represents the difference between the output $y_1(k)$ of the first microphone **115** and the output $\hat{y}_1(k)$ of the adaptive filter **135** and is referred to herein as an error signal.

In operation, filter coefficients of the adaptive filter **135** are adjusted using a least-squares algorithm such that the error signal e(k) is minimized. In other words, the adaptive filter **135** is adjusted such that the output $\hat{y}_1(k)$ of the adaptive filter **135** is as close as possible to (i.e., is an estimator of) the output $y_1(k)$ of the first microphone **115**. Thus, the adaptive filter **135** attempts to model the signal effects created by the physical separation of the microphones **115**, **125**. For example, when the passenger **120** is speaking, his or her voice will reach the first microphone **115** slightly later than it will reach the second microphone **125**, and the corresponding speech signal level received at the first microphone **115** will be somewhat attenuated as compared to the level received at the second microphone **125**. Thus, the adaptive filter **135** is adjusted to provide similar delay and attenuation effects.

As a result, the relative time delay and signal attenuation at the microphones with respect to each user can be calculated based on the coefficients of the adaptive filter **135** as described, for example, in Y. T. Chan, J. M. Riley and J. B. Plant, "A parameter estimation approach to time delay estimation and signal detection", IEEE Transactions on Acoustics, Speech and Signal Processing, vol. ASSP-28, February, 1980, which is incorporated herein in its entirety by reference. One disadvantage of the system of FIG. **1**, however, is that its performance deteriorates significantly in the presence of background noise. As a result, the system of FIG. **1** is not useful in most practical applications, where significant background noise (e.g., road and traffic noise) is commonplace. Thus, there is a need for improved methods and apparatus for measuring relative signal levels and time delays at multiple sensors.

SUMMARY OF THE INVENTION

The present invention fulfills the above-described and other needs by providing a system in which a fixed filter and an adaptive filter are used in combination to provide accurate and robust estimates of signal levels and time delays for multiple sensors. In exemplary embodiments, the fixed filter includes at least one relatively narrow passband which is used to distinguish signal sources of interest from broadband background noise. In the embodiments, the fixed filter is coupled to a reference sensor and the adaptive filter is coupled to a secondary sensor. An error signal derived from the outputs of the fixed filter and the adaptive filter is used to adjust filter coefficients of the adaptive filter according to a suitable least-squares algorithm. The coefficients of the fixed filter and the adaptive filter are used to compute estimates of the time delay and relative level between the two sensors. The estimates can then be used to make decisions regarding sensor selection and beamforming.

In exemplary embodiments, the functionality of the system is supplemented with an activity detector which indicates when no signal of interest is present. In the activity detector, accumulated energy in the adaptive filter is compared with an expected least value derived from the coefficients of the fixed filter. When the accumulated energy is smaller than the expected value, indicating that there is no signal of interest present (i.e., only background noise is present), the time delay and relative level estimates are set to appropriate values to ensure proper operation of the system even during periods where no signals of interest are present.

In additional embodiments, more than two signal sensors are employed. In such embodiments, one sensor is treated as a reference sensor and coupled to a fixed filter, while each of the additional sensors is coupled to an adaptive filter. For each additional sensor, an error signal derived from the outputs of the fixed filter and the corresponding adaptive filter is used to update the coefficients of the corresponding adaptive filter. Thus, robust estimates of the time delay and relative signal level between the reference sensor and each additional sensor can be computed, and sophisticated decisions can be made with respect sensor selection and beamforming.

Generally, the present invention provides a computationally simple yet accurate and robust method for estimating the time delays and relative signal levels at multiple sensors. The teachings of the invention are applicable in a wide variety of signal processing contexts. For example, in addition to the hands-free mobile telephony application described above, the invention may be used for other acous-